IN THE CLAIMS

1. - 3. (canceled)

4. (currently amended) ~~The end-to-end bandwidth estimation according to claim 2~~ In an end-to-end estimation method of bandwidth available in a connection of a client and server established over a packet switching network, the improvements comprising:

computing in a first routine samples of the bandwidth available by taking into account a flow of data packets received by the client and time intervals during which the data packets are received if the routine is implemented at a receiver side of the client, or by taking into account acknowledgements or report packets received by a sender side of the server and time intervals during which acknowledgment or report packets are received if the routine is implemented at the sender side of the server;

computing in a second routine samples of available bandwidth as a ratio of an amount of received data packets over the time intervals during which the data packets are received if the routine is implemented at the receiver side of the client, or as a ratio of an amount of the data packets acknowledged over the time intervals during which the data packets are acknowledged if the routine is implemented at the sender side of the server; and

implementing in a third routine a discrete time low-pass filter to obtain a filtered value of the samples of the available bandwidth,

wherein a sample of the bandwidth available b_j at a time t_j is computed as:

$$b_j = \frac{d_j}{t_j - t_{j-1}}$$

where d_j is the amount of the received data packets that have been received at the receiver side of the client or acknowledged at the sender side of the server in an interval $t_j - t_{j-1}$, t_{j-1} is a time when a previous ACK or an ACK of one or more congestion windows of packets before were received by the sender side of the server or a time when a previous packet or a packet one or more congestion windows of packets before were received by the receiver side of the client, and t_j is a time when a current ACK is received by the sender side of the server or when a current packet is received by the receiver side of the client

, and

wherein the available bandwidth samples are ~~computed according to claim 2 and are~~ averaged using the discrete-time low-pass filter with time-varying coefficients:

$$\hat{b}_j = \frac{2\tau_f - \Delta_j}{2\tau_f + \Delta_j}\hat{b}_{j-1} + \Delta_j \frac{b_j + b_{j-1}}{2\tau_f + \Delta_j}$$

where $\hat{b}_j$ is the filtered measurement of the available bandwidth at time $t = t_j$, $\hat{b}_{j-1}$ is the filtered measurement of the available bandwidth at time t_{j-1}, $\Delta_j = t_j - t_{j-1}$, $1/\tau_f$ is the cut-off frequency of the filter, b_j is the sample of the available bandwidth at time t_j, and b_{j-1} is the sample of the available bandwidth at time t_{j-1} if a time τ_f/m ($m \geq 2$) has elapsed since the last received ACK or packet without receiving any new ACK or packet, then the filter assumes the reception of a virtual sample b_j=0.

5. - 6. (canceled)

7. (currently amended) Method for adaptively setting congestion window (cwnd) and slow start threshold (ssthresh) in the TCP/IP protocol comprising ~~an end-to-end bandwidth estimation according to claim 1~~ an end-to-end estimation method of bandwidth available in a connection of a client and server established over a packet switching network, the improvements comprising:

computing in a first routine samples of the bandwidth available by taking into account a flow of data packets received by the client and time intervals during which the data packets are received if the routine is implemented at a receiver side of the client, or by taking into account acknowledgments or report packets received by a sender side of the server and time intervals during which acknowledgment or report packets are received if the routine is implemented at the sender side of the server;

computing in a second routine samples of available bandwidth as a ratio of an amount of received data packets over the time intervals during which the data packets are received if the routine is implemented at the receiver side of the client, or as a ratio of an amount of the data packets acknowledged over the time intervals during which the data packets are acknowledged if the routine is implemented at the sender side of the server; and

implementing in a third routine a discrete time low-pass filter to obtain a filtered value of the samples of the available bandwidth to set the windows as follows:

after a timeout: set ssthresh=min(2, BWE*RTTmin)

set cwnd=2;

after 3 dupack: set ssthresh=min(2, BWE*RTTmin)

set cwnd=ssthresh; and

wherein RTT min is the minimum round trip time and BWE is the available bandwidth computed ~~according to claim 1~~ at the time of timeout or when 3 dupacks or n are received.

8. (canceled)

9. (currently amended) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol, comprising:

a routine to compute an end-to-end estimation of the available bandwidth according to claim 7;

a routine that selects the quality of coding, or the nuembers of layers to be transmitted in a layered coding of an audio/video so that the sending rate is the closer to the end-to-end bandwidth available estimated according to claim 7; and

a routine to set TCP congestion window and slow start threshold according to claim 7 in order to send the coded audio/video source.

10. - 12. (canceled)

13. (previously presented) Method for adapting the amount of data for unit of time sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low pass filter is a low pass filter according to claim 4.

14. (previously presented) Method for adaptively setting congestion window and slow start threshold in the TCP/IP protocol comprising an end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4.

15. (previously presented) Method for adaptively selecting the quality of coding, or the numbers of layers to be transmitted in a layered coding of an audio/video source using the TCP protocol, or the UDP protocol or the RTP protocol comprising and end-to-end bandwidth estimation comprising:

a routine to compute samples of available bandwidth by taking into account packets received by the client, if the routine is implemented at the receiver side, or by taking into account acknowledgment packets received by the sender, if the routine is implemented at the sender side; and

a routine that implements a discrete time low-pass filter to obtain a filtered value of the samples of available bandwidth over a packet network, comprising an end-to-end bandwidth,

wherein the low-pass filter is a low-pass filter according to claim 4,

16. - 19. (canceled)

20. (previously presented) Method for adapting the amount of data for unit of time, i.e. the rate, sent by the server to the client over a packet network, comprising an end-to-end bandwidth estimation according to claim 4.

21. (canceled)